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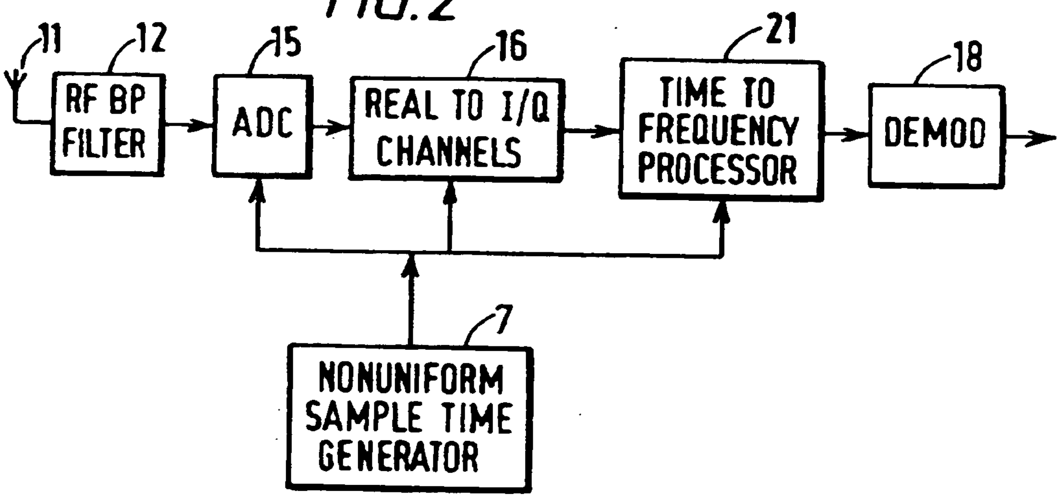
(54) Abstract Title
OFDM Demodulation

(57) An OFDM receiver demodulates the received signal by sampling at non-periodic intervals, and converting the sampled signal into the frequency domain. The RF signal can be sampled to avoid down-converting to the IF frequency, or the IF signal can be sampled. Several ways of determining the distribution of the carriers required for demodulation are described.

FIG. 2



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At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.

FIG. 1

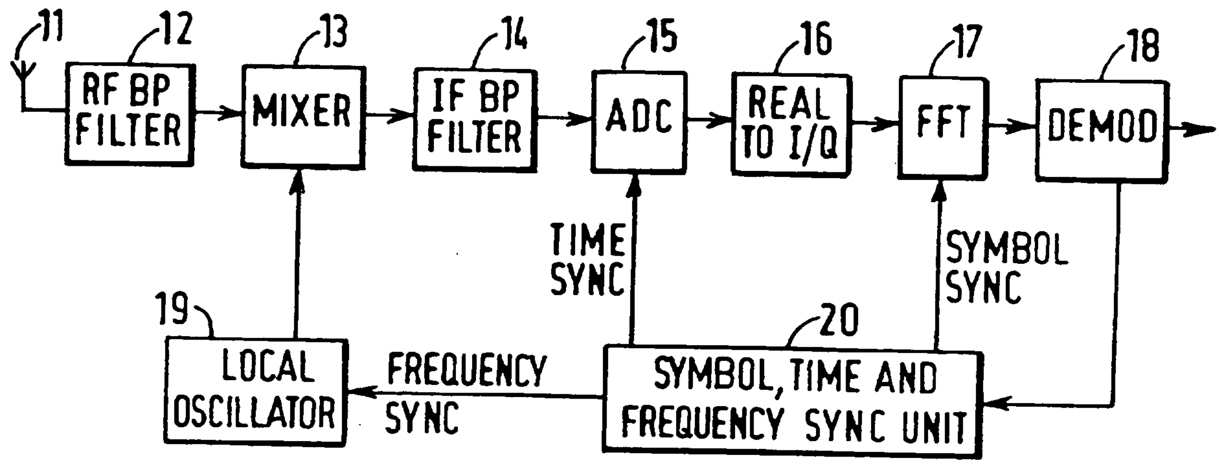


FIG. 2

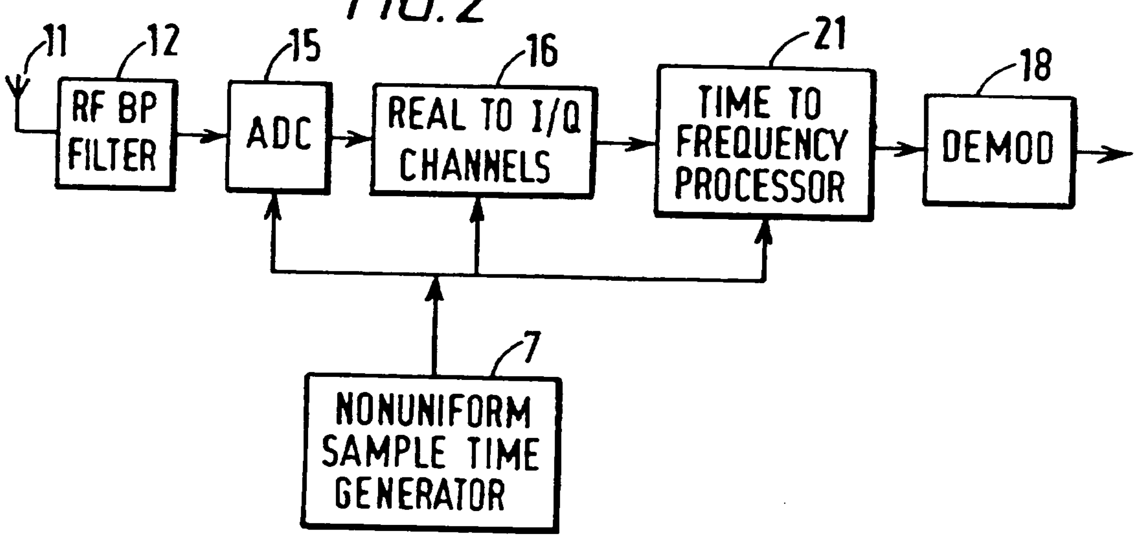


FIG. 3

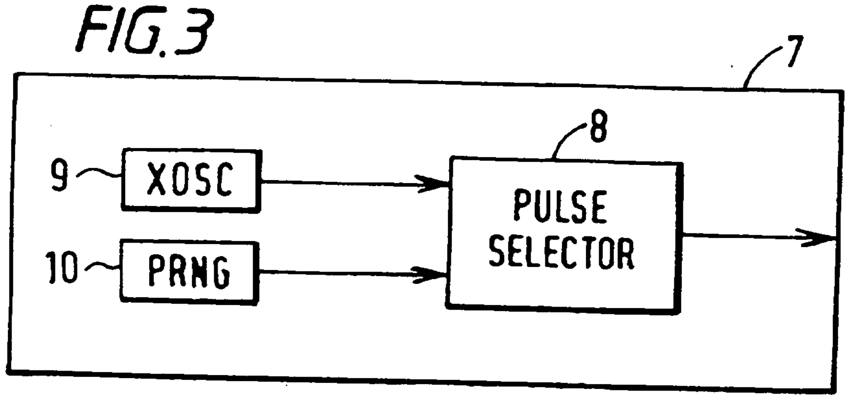


FIG.4

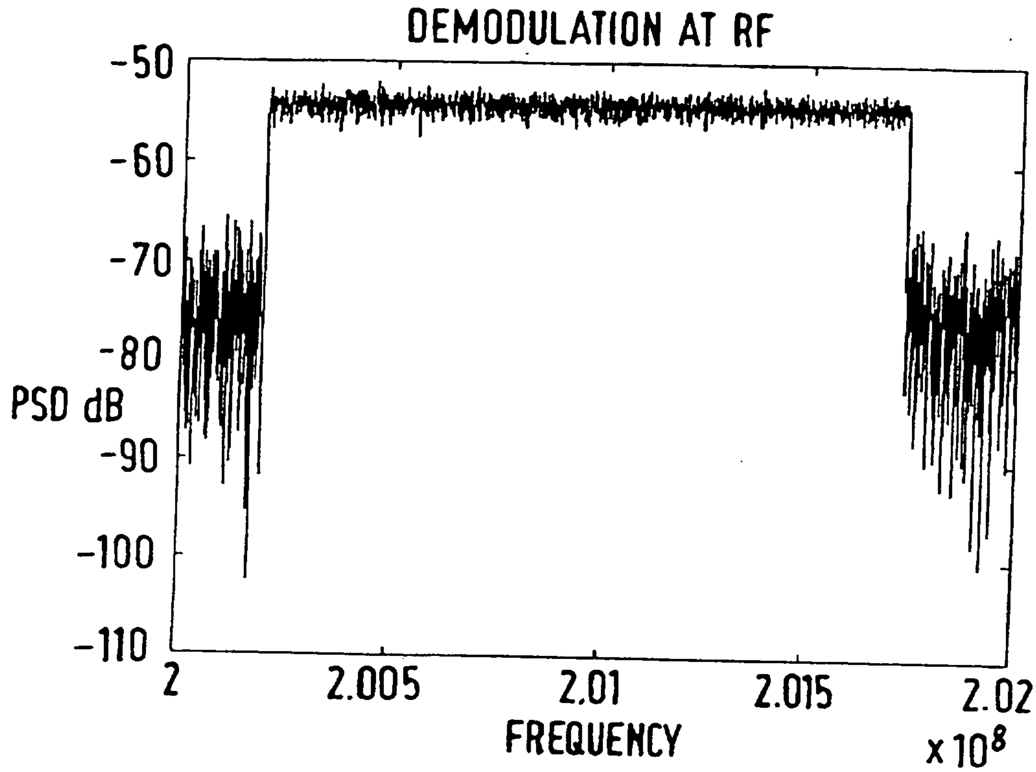

DEMODULATION AT RF
-50
-60
-70
-80
PSD dB
-90
-100
-110
2
2.005
2.01
2.015
2.02
FREQUENCY
x 10^8

FIG.5

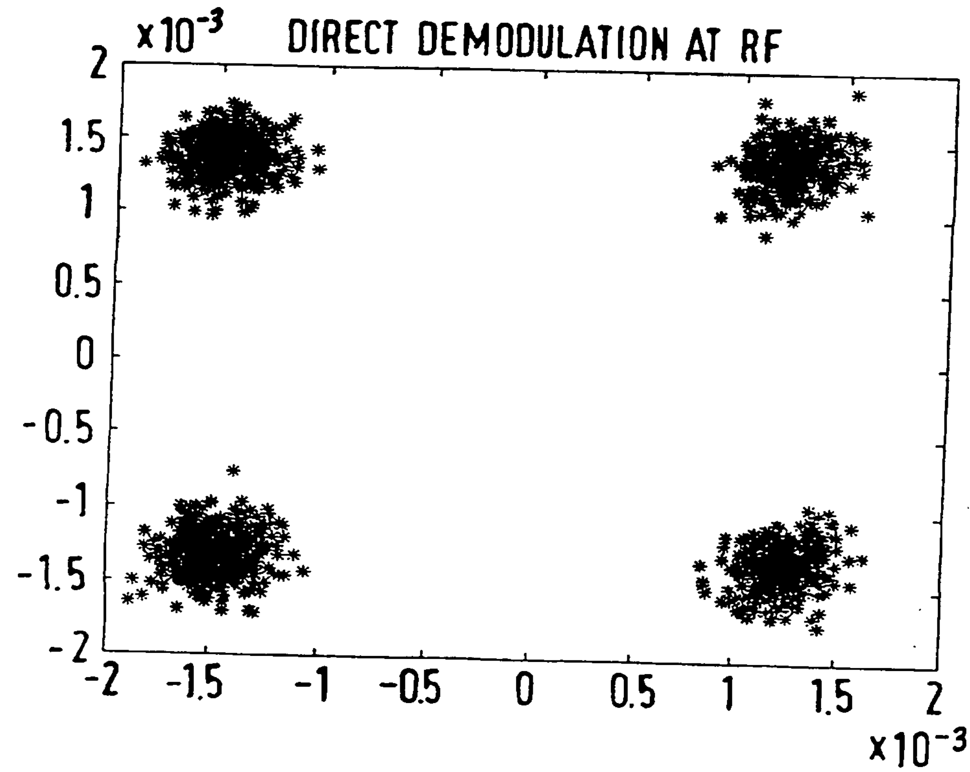

x 10^-3
DIRECT DEMODULATION AT RF
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
-2
-1.5
-1
-0.5
0
0.5
1
1.5
2
x 10^-3

FIG.6

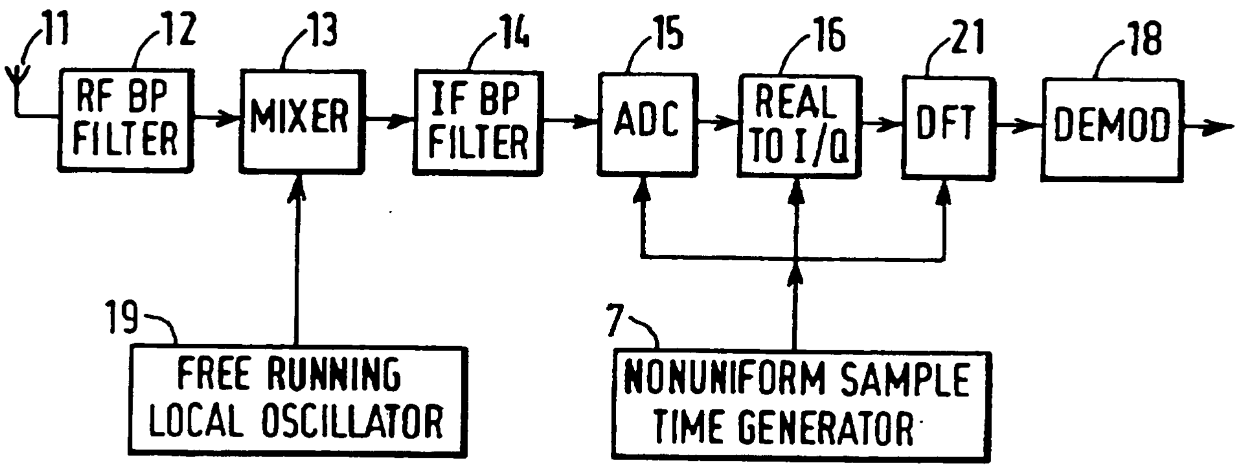
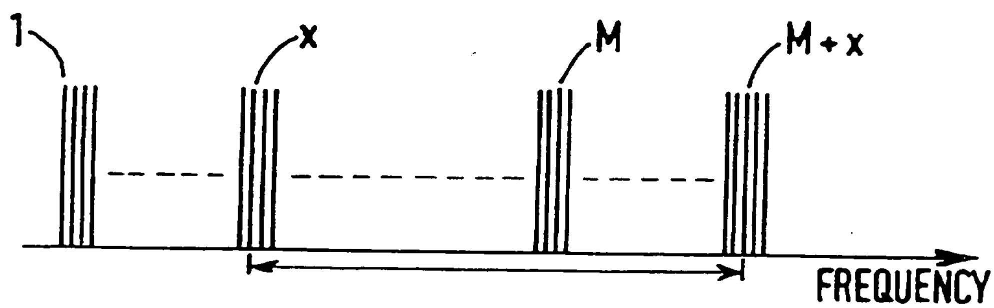

11
12
RF BP FILTER
13
MIXER
14
IF BP FILTER
15
ADC
16
REAL TO I/Q
21
DFT
18
DEMOD
19
FREE RUNNING LOCAL OSCILLATOR
7
NONUNIFORM SAMPLE TIME GENERATOR

FIG.7


1
x
M
M+x
FREQUENCY

2344729

OFDM Demodulation

This invention relates to Orthogonal Frequency Division Multiplexing (OFDM), and particularly to the demodulation of OFDM signals. OFDM signals, in which transmitted information is modulated in parallel onto a large number of precisely-located adjacent carriers, has a number of well known advantages, including avoidance or mitigation of fading due to multi-path interference. This is particularly advantageous for the use of audio broadcasting (DAB), and also for the use of video broadcasting (DVB).

The conventional approach to demodulation of an OFDM signal involves generating an accurate local oscillator signal which is mixed with a received RF signal. This produces an intermediate frequency signal which is then sampled. The samples are then converted into separate real and imaginary components, which are transformed into the frequency domain. This provides multiple outputs each associated with a respective carrier (in one exemplary DAB signal there would be 1,536 carriers), each output representing one of four possible phase values.

Any errors in the frequency of the local oscillator will result in sampling errors. Accordingly, much effort has gone into techniques for improving the accuracy and speed of the locking-in process during tuning. In addition, effort has also gone into ensuring that the sampling times of the analogue-to-digital converter are exactly aligned to the received signal, and that

the Fourier transform process which is used for converting into the frequency domain is accurately aligned to the received symbols.

It would be desirable to mitigate the problems inherent in meeting these requirements, and also to provide a simpler and less expensive demodulator.

According to one aspect of the invention, an OFDM signal is demodulated by sampling a received signal (either RF or IF) at random intervals, the sample signals then being converted into the frequency domain. It has been found that random sampling, and particularly non-periodic or substantially non-periodic sampling, permits the modulated carriers to be recovered, but avoids the need for a local oscillator (if the sampling is used for directly extracting the demodulated signal from the RF signal), or for an accurate local oscillator (if the IF signal is sampled). This technique also avoids the need for time synchronisation and only requires approximate symbol synchronisation.

It will therefore be appreciated that this aspect of the invention departs significantly from the conventional approach of ensuring precise tuning to the received signal, and subsequently sampling the signal at precise, predetermined instants, by instead performing sampling at random intervals.

Preferably, the sampling times form an additive random sequence of the form $t_n = t_{n-1}+e_n$.

where t_n is the n^{th} sampling time, t_{n-1} is the preceding sampling time and e_n is a period whose value changes in a random manner for each interval. This

type of sequence is less periodic, and thus preferable, to other sequences such as those used for conventional dither techniques in which the sample time $t_n=nT+e_n$, i.e. the sampling interval is equal to a nominal interval T plus a randomly determined period. Any periodicity in the signal results in aliasing, and limits the frequency range over which the technique is effective.

In the arrangement embodying the invention will now be described by way of example with reference to the accompanying drawings, in which:

Figure 1 is a schematic block diagram of a conventional OFDM receiver;

Figure 2 is a block diagram of an embodiment of a receiver in accordance with the present invention;

Figure 3 is a block diagram of a sample time generator for use in the embodiment of Figure 2;

Figure 4 is a spectrograph of an RF signal as presented to a demodulator of the embodiment;

Figure 5 is a scatter diagram, plotted with an arbitrary scale, showing the phases of the demodulated signal;

Figure 6 is a block diagram of a second embodiment of a receiver in accordance with the invention; and

Figure 7 schematically illustrates the output of a time to frequency converter of the embodiment.

Referring to Figure 1, a conventional OFDM receiver includes an antenna 11 for receiving a transmitted RF signal. An RF bandpass filter 12 allows the chosen multiplex signal to pass, and removes other signals. A mixer 13 receives the RF signal as well as the signal from a local oscillator 19 and produces the sum of and difference between the RF signal and the output of the local oscillator. An IF bandpass filter 14 attenuates the unwanted sideband signal, and provides an IF signal to an analogue-to-digital converter (ADC) 15. The ADC 15 samples the filtered IF signal at precise, regular intervals. The sampled signal is then converted, by converter 16, from a real signal into in-phase and quadrature components, referred to as I and Q signals.

A processor 17 performs a fast Fourier transform (FFT) to convert the I and Q signals into the frequency domain.

A phase demodulator 18 extracts the phase of the FFT output which contains the transmitted data. Thus, for each carrier, the Fourier transform of the I and Q signals produces phase information representing the modulation symbol for that carrier.

A symbol, time and frequency synchronisation unit 20 receives signals from the phase demodulator 18 and generates (a) a frequency synchronising signal which is used to control the local oscillator 19 to ensure that the receiver is precisely tuned, (b) a time synchronisation signal which is fed to the ADC 15 to ensure sampling at the correct instants, and (c) a symbol synchronisation

signal which is delivered to the FFT processor 17 to ensure that the start of each transmitted symbol is located correctly.

Referring to Figure 2, a first embodiment of the present invention is shown herein. Components corresponding to those in Figure 1 have the same reference numbers.

It will be noted that there is no local oscillator, mixer or intermediate frequency filter. Instead, the filtered RF signal is delivered straight to the ADC 15. This samples the signal at instants determined by a non-uniform sample time generator 7, shown in more detail in Figure 3. Sample time generator 7 comprises a crystal oscillator 9, the output pulses of which are sent to a pulse selector 8. The pulse selector also receives the output of a pseudo-random number generator 10, which constitutes an index determining which pulses to delete and which to leave in the pulse stream. In one example, the interval between successive selected pulses p_n and p_{n-1} corresponds to r_n clock pulses, where r_n is the generated random number. The selected pulses are output from the generator 7. The output is thus a stream of pulses with a random integer number of clock periods between each pulse. (The term "random" sequence as used herein is intended to cover pseudo-random sequences.)

The output of the ADC 15 is then delivered to an I/Q converter 4 which uses the sample times to perform the conversion, and the I and Q signals are then converted into the frequency domain by a time-to-frequency converter 21. In this embodiment, the standard FFT processor cannot be used, because this

relies on the received samples being derived at regular intervals. Accordingly, a discrete Fourier transform (DFT) processor is used as the converter 21. The sample times are delivered to and stored by the DFT processor 21 so that they can be used in the conversion process.

The output of the processor 21 is delivered to a demodulator 18.

Figure 4 is a spectrograph showing the output power of the DFT processor 21, as measured over a symbol period, in one particular example. It is evident that the random sampling causes smeared alias frequencies which appear to look like noise. It would be possible to reduce this by increasing the average sampling rate. However, even with the noise level shown in Figure 4, the central, high plateau region of the chart representing the region containing the carriers is easily distinguishable from the remaining areas. There is approximately a 20 dB separation between the received carriers and the levels outside the frequency range of interest.

Figure 5 illustrates the effect of the alias frequencies on the demodulation process. In Figure 5, each point represents the detected phase of an individual modulation symbol associated with a respective carrier. In an ideal, perfectly tuned system, the chart would show a single point value in each of the four quadrants. However, because of the random sampling, the values in each quadrant are smeared. Nevertheless, symbols can be demodulated without error, so long as the groups of points do not overlap.

Figures 4 and 5 represent a particular example involving simulating a DAB 1536 carrier signal, with a centre frequency of 201MHz, 1kHz carrier spacing, 1ms symbol duration and a random phase on each carrier. To extract the symbols, 8000 non-periodic samples are taken at an average frequency of 8MHz. A suitable sampling function for a DAB signal is an additive random time sequence with a mean sampling frequency of 8MHz, a standard deviation of 0.3 and a largest common divisor of 2ns which is 1/500MHz. The figures represent the output of the 2000 point complex DFT (assuming 8 bits for real and 8 bits for imaginary sampled data).

Similar results can be expected in practical embodiments, for example a receiver suitable for the Eureka 147 DAB system operating in mode I. The parameters for this signal are 1536 carriers, spaced 1kHz apart, with a symbol duration of 1246μs and a guard interval of 246μs. The UK RF DAB frequencies are in the region of 220MHz. For these parameters a suitable crystal oscillator frequency is in the region of 500MHz. The RF filter 12 preferably has a bandwidth of approximately 2MHz.

It is to be noted that the RF filter 12 is preferably of narrow bandwidth to avoid any large adjacent signals producing a smeared aliasing which would drown the wanted signal.

An alternative embodiment is shown in Figure 6. In this case, the RF signal from the filter 12 is converted into an IF signal, and the filter 12 is not required to have such a narrow and precisely-located bandwidth. The mixer 13

receives a signal from a free-running local oscillator 19 to produce the IF signal which is filtered by the IF filter 14. The ADC 15 therefore samples an IF signal, rather than an RF signal as in Figure 2.

Because the local oscillator 19 is free-running, the IF carrier frequencies are not exactly predictable. However, assuming that the number of complex point Fourier transforms exceeds the number of carriers (for example there may be a 2000 complex point Fourier transform and 1536 carriers), the carriers will still be within the frequency range of the transform.

It will be appreciated that the locations of the transformed carriers within the output of the DFT may vary depending upon the local oscillator frequency. Referring to Figure 7, the vertical lines represent the "frequency bins" of the output of the DFT processor. In conventional receivers, the local oscillator frequency is adjusted so that the M carriers are located in predetermined bins, e.g. bins I to M. The demodulator then generates data based on the contents of those bins. In the present embodiment, the local oscillator is not adjusted. Accordingly, the distribution of the carriers is determined (e.g. the offset x representing the position of the lowest frequency carrier) and the demodulator then outputs data based on the contents of the carriers in bins x to M + x. The frequency spectrum over which the transformed carriers are distributed can be determined in any of a number of ways. For example, each of the frequency outputs of the Fourier transform process can be checked against a threshold to determine those frequency slots

containing transformed carriers. Alternatively, the demodulated data for each frequency slot can be examined. Those frequency slots which contain randomly distributed phases are assumed to not contain any carriers, whereas those frequency slots in which the phases are substantially equal to the four expected phases corresponding to signal modulations are assumed to be carrier-containing frequency slots. As a further alternative, the distribution of the transformed carriers can be determined by examining the content of the demodulated data. In one specific example, the output of the demodulator is checked to determine whether the pattern of individual modulation symbols corresponds to a known transmitted pattern, for example the pattern of the phase reference symbols transmitted in conventional systems.

The demodulator can therefore determine the location of the OFDM carrier within the output spectrum of the DFT processor, and use this information as indicated above in generating the demodulated output data.

It will be understood that by using these techniques there is no need to ensure that individual carriers are transformed into particular frequency slots as in the prior art, because any variation in the distribution of the carriers can be corrected. Because of this correction, certain types of signal distortion resulting in a spectrum shift of the received RF signal (for example a Doppler shift) are also corrected. These distribution correction techniques can therefore also be applied to advantage in the embodiment of Figure 2, to deal with such signal distortions.

Instead of using a free-running oscillator, the local oscillator 19 could be locked to a stable external source to reduce frequency drift over time, or it could be locked to the incoming received signal as in a conventional receiver, to compensate for any frequency shift.

CLAIMS:

1. A method of demodulating an OFDM signal, comprising sampling the signal at substantially non-periodic intervals and converting the sampled signal into the frequency domain.

2. A method as claimed in claim 1, wherein the sampled signal is an RF signal.

3. A method as claimed in claim 1, wherein the sampled signal is an IF signal.

4. A method as claimed in claim 3, wherein the IF signal is derived by mixing an RF signal with a free-running local oscillator signal.

5. A method as claimed in claim 3, wherein the IF signal is derived by mixing an RF signal with a stabilised local oscillator signal.

6. A method as claimed in claim 3, wherein the IF signal is derived by mixing an RF signal with a local oscillator signal locked to the received signal.

7. A method as claimed in any preceding claim, wherein the sampling times form an additive random sequence.

8. A method as claimed in any preceding claim, the method comprising determining the distribution of the OFDM carrier within the spectrum of the signal transformed into the frequency domain, and generating output data taking that distribution into account.

9. A method of demodulating an OFDM signal, the method being substantially as herein described with reference to the accompanying drawings.

10. Apparatus for demodulating an OFDM signal, the apparatus operating according to a method as claimed in any preceding claim.

INVESTOR IN PEOPLE

Application No: GB 9826884.0
Claims searched: 1 to 10

Examiner: Ken Long
Date of search: 10 June 1999

Patents Act 1977
Search Report under Section 17

Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.Q): H4P (PAL, PAQ & PAPD)

Int Cl (Ed.6): H04L 27/26, 27/227 & 27/233
H04J 11/00

Other: ONLINE : EPODOC, WPI, JAPIO

Documents considered to be relevant:

Category	Identity of document and relevant passage		Relevant to claims
A	EP 0836304 A2	ALPINE	None
A	US 5471464	SONY	None
A	US 5357502	FRANCE TELECOM	None

X Document indicating lack of novelty or inventive step
Y Document indicating lack of inventive step if combined with one or more other documents of same category.
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E Patent document published on or after, but with priority date earlier than, the filing date of this application.